REMARKS

By this Amendment, claim 1 has been canceled without prejudice and has been replaced by new claim 60, and claims 2-6, 8, 11-14, 16-21, 27, 29-37, 39-44 and 48 have been amended. Thus, claims 2-60 are presently pending in this application. Of these, the Examiner has withdrawn claims 23-26 and 56-59 from consideration. It is noted that claims 56-59 have been withdrawn from consideration because these claims are dependent on withdrawn claim 26. It is further noted that claim 28 is dependent on withdrawn claim 23. It is accordingly assumed that claim 28 should also be deemed withdrawn from consideration, and will be so treated herein.

Applicant notes for the record, that claims 2-6, 8, 11-14, 16-21, 27, 29-37, 39-44 and 48 have been amended to change the dependency thereof from canceled claim 1 to new claim 60, and to replace the word "said" by -- the -- in several claims in which "said" and "the" were both words used as definite articles. The scope of these claims has not been narrowed by any of the changes. Similarly, claim 1 was rewritten as claim 60 to better emphasize previously recited features of claim 1, to clarify the preamble, and to improve the language by which the various features are recited. This claim is, however, of the same scope as original claim 1.

Applicant respectfully requests reconsideration and withdrawal of the outstanding rejection of claims 1-22, 27, and 29-55 as anticipated by Kenyon et al., U.S. Patent No. 4,450,531, or as obvious over Kenyon et al. in view of Uehara, U.S. Patent No. 5,754,798 or Hoffberg et al. U.S. Patent 5,901,246.

Preliminarily, it is noted that claims 30-53 are not mentioned in the first sentence of Section 6 of the Office Action, but from the rest of the section, it appears that the Examiner's intention was to include these claim in the rejection stated therein. For purposes of this response, applicant assumes that claims 30-53 also stand rejected as anticipated by Kenyon et al.

Further, in the first sentence of Section 9 of the Office Action, claims 22 and 55 are stated to be rejected as unpatentable over Kenyon et al. in view of Uehara, but in the remainder of the section, the Hoffberg et al. patent is applied as the secondary reference, and not Uehara. It is therefore assumed that Hoffberg et al. is the intended secondary reference.

Turning now to the merits of the rejections, Kenyon et al., like the present invention, does relate to broadcast program surveying and does disclose use of signal processing techniques, but these are

superficial similarities. What is taught in the Kenyon et al. patent is quite different in both concept and execution from the present invention.

To begin with, Kenyon et al. are not concerned with creating an ultra-compact storage device which can conveniently be *worn* by an individual whose listening behavior is being surveyed, but rather with monitoring the behavior of a broadcaster without the need for devoting bandwidth to special coding of program material, and with overcoming monitoring unreliability due to signal dropout or variable speed playback of music by disk jockies. Short sampling times and extremely high compression are also not of concern to Kenyon et al. (see column 2, lines 44-54).

Further, neither the signals recorded, nor the way the recorded signals are processed according to Kenyon et al. are the same as in the present invention.

Consider first, claim 60 which has been substituted for original claim 1. This claim is directed to "[a] method for storing an electric signal representing recorded ambient noise in compressed form". Kenyon et al. in contrast, is not concerned with signals representing ambient noise, but instead, with samples taken directly from the broadcast signal (see column 4, lines 21-22), which are compared with reference samples taken at the broadcast station. Ambient noise, with which the present invention is concerned, is of strongly varying intensity and represents an undefined mixture of noise sources, one of which may be the one of several programs to be surveyed.

The differences in basic concept between Kenyon et al. and the present invention are clearly reflected in the method of claim 60, which comprises the step of:

> periodically recording samples of the ambient noise using a sound transducer. . .

In Kenyon et al. there is no periodic sampling of ambient noise, as mentioned above. Further, claim 60 calls for the step of:

> normalizing the amplitude of a signal output of the transducer within a first predetermined range D . . .

The normalization described in Kenyon et al. at column 4, lines 36 - 52 (quoted by Examiner) relates to the reference samples, and not to the broadcast or "real" samples, and thus can not even be analogized to applicant's ambient noise signals. Also, the reference samples in Kenyon et al. are

frequency filtered, digitized, and then Fourier transformed. Hence, Kenyon et al. do not normalize the samples, even of the reference signals.

In fact, it is not even the *amplitude* of the reference samples which is normalized according to Kenyon et al., but rather the energy or power (see column 4, lines 42-49). Such a normalization does not prevent the amplitude from locally exceeding a certain upper limit. According to the present invention, the sampled ambient noise amplitude is normalized so that the highest peak value within the sample matches with the value D, i.e., the maximal value of the range from O to D.

Claim 60 also calls for:

> mapping the normalized amplitude values of the sampled ambient noise onto a second predetermined range of values using a non-linear mapping function to obtain an emphasis of selected value ranges within the first and/or second predetermined ranges . . .

According to the present invention, the normalized amplitude values are *mapped* to a second range of values using a nonlinear function. As a result, values in part of the input range D are spread, i.e. emphasized, while values in other parts of input range D are compressed, i.e. "weakened". In the preferred execution, the small values are spread or emphasized and the large values are compressed. An exemplary non-linear mapping function to produce this result might be W = log (D).

In contrast, Kenyon et al. perform a Fourier transformation, not a mapping by a function. A Fourier transformation is not a mapping, in that the transformation takes into account all values within a certain interval for the calculation of each of the set of result values. That means that there is no exact relation between one particular input (amplitude) value and a particular output value. With a mapping such as W = log (D), there is a one-to-one correspondence between each output value and a particular input value.

Nor is the rejection supported by the Examiner's reference to Fig. 1 of the Kenyon et al. patent. Perhaps the Examiner has given significance to the fact that Fig. 1 of the patent refers to ranges W and D. This correspondence is purely coincidental. These ranges of Kenyon are abcsissa value ranges, i.e., they constitute points in time. The range D and W according to the present invention denote *amplitude* values, or values derived therefrom.

Finally, claim 60 calls for the step of:

> storing the mapped result in an electronic memory in a digital format.

As to this, it is once again noted that in Kenyon et al., the processed *reference* samples are stored for later use. As explained above, these are not mapped, normalized ambient noise amplitude samples as in the present invention. Nor are any processed and stored signals derived from the sampled broadcast signals. This is clear from column 5, lines 62 - 64 in which its is stated that the reference samples are stored so that they ". . . can be provided at various locations for comparison with a broadcast signal at that location."

From the foregoing, it may be seen that none of the recited features of claim 60 are taught or suggested in Kenyon et al.

These deficiencies are not remedied by anything taught or suggested in the Uehara or Hoffberg et al. patents. Neither Uehara nor Hoffberg et al. has anything at all to do with broadcast program monitoring. Even if it could be said to be obvious to modify Kenyon et al. according to anything taught or suggested in either of the secondary references, the result would still not satisfy the terms of independent claim 60. It is therefore respectfully submitted that claim 60 should be allowed.

Claims 2-22, 27, and 29-55 are all dependent on allowable claim 60, and should be allowed for all the reasons stated above. In addition, these claims recite features which, in combination with the features of claim 60, are not taught or suggested in any of the prior art, considered either alone or together.

Finally, claims 23-26, 28, and 56-59 were withdrawn from consideration on the basis that linking claim 1 had not been allowed. Since claim 60, which has replaced claim 1 has been demonstrated to be allowable, it is respectfully submitted that the withdrawn claims should now be examined and allowed for the reasons stated in connection with base claim 60.

In view of the foregoing, favorable reconsideration and allowance of this application are respectfully solicited.

I hereby certify that this correspondence is being deposited with the United States Postal Service with sufficient postage as First Class Mail in an envelope addressed to: Asst. Commissioner for Patents, Washington, D.C. 20231, on June 11, 2002

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APPENDIX A

"CLEAN" VERSION OF EACH PARAGRAPH/SECTION/CLAIM

37 C.F.R. § 1.121(b)(ii) AND (c)(i)

SPECIFICATION

Paragraph beginning at page 2, line 24 **(Delete in its entirety)**

CLAIMS (with indication of amended or new):

Sub F1

2. (Twice Amended) The method of claim 60, wherein said nonlinear function has a slope dW/dD which decreases with increasing values in the range D to obtain an emphasis of the smallest values of the first range of values.

3. (Twice Amended) The method of claim 60, wherein the mapped result is represented by binary numbers having a fixed number of binary digits from 3 to 16 bits.

Sub E2 C1

4. (Twice Amended) The method of claim 60, further comprising dividing the audio signal into at least two band signals by filtering, with each one of the band signals containing a frequency range of the audio signal, and wherein any content of the other band signals contained in each band signal is present only in an attenuated form.

Sub F1

5. (Twice Amended) The method of claim 4, wherein the audio signal is divided into from 3 to 15 band signals.

C2 Sub

6. (Amended) The method of claim 4, wherein the band signals essentially contain frequency ranges of the same width each, and all frequency ranges are comprised in the range of 500 Hz to 10,000 Hz.

C3

8. (Twice Amended) The method of claim 7, wherein the low pass filtering is realized by means of a digital convolution over 10-30 values.

C4

11. (Twice Amended) The method of claim 60, further comprising generating an energy signal which is proportional to an energy content of the ambient noise from the audio signal or from a signal derived from the audio signal.

C5

12. (Amended) The method of claim 11, wherein the energy signal is subjected to a second low pass filtering.

13. (Twice Amended) The method of claim 12, wherein the second low pass filtering is effected digitally in the form of a convolution over 20 to 70 values.

C6

14. (Twice Amended) The method of claim 13, wherein the second low pass filtering is followed by a second data reduction where one energy value among n filtered values is selected, n being at least equal to 2.

Sub E3

C7

16. (Amended Three Times) The method of claim 60, wherein the range of normalized values D is defined by a lower limit D_u, and an upper limit D_o, and wherein the normalization is effected by:

- obtaining the maximum of the absolute value of the audio signal or the derived signal within the duration of normalizing the audio or derived signal, which is shorter than or equal to the duration of a hearing sample,
- multiplying the reciprocal value of said maximum by $(D_0 - D_u+1)$, and
- multiplying this product by each value of the audio or derived signal within the duration of the normalized signal.

17. (Twice Amended) The method of claim 60, wherein essentially all steps of the method are performed by integer or fixed point arithmetic.

C8

18. (Twice Amended) Device for carrying out the method of claim 60, comprising a hearing sample unit comprising at least one signal processor for performing at least one processing step of the method.

19. (Twice Amended) The device of claim 18, further comprising a non-volatile semiconductor memory connected to the processor for storing the results of the method.

20. (Twice Amended) The device of claim 18, further comprising a timer connected to a power supply of the hearing sample unit for switching off the hearing sample unit when no processing activity is required.

21. (Twice Amended) The device of claim 19, wherein a power supply of said non-volatile memory and/or the memory itself is connected to a timer in such a manner that the memory is essentially capable of being operated only during the storage of the results in order to reduce the energy consumption by the memory.

C9

27. (Twice Amended) A magnetic, optical or magneto-optical data carrier, containing a recorded program which executes the method according to claim 60.

C10

29. (Twice Amended) Device comprising at least one program controlled processor unit and a memory for storing a program controlling the processor unit, wherein the memory contains a program which controls at least one of the operations of the method of claim 60.

30. (Amended) The method of claim 60, wherein the electroacoustic transducer is a microphone.

31. (Amended) The method of claim 3, wherein the mapped result is represented by binary numbers having a fixed number of binary digits from 4 to 8 bits .

32. (Amended) The method of claim 3, wherein the mapped result is represented by binary numbers having 4 bits of binary digits.

33. (Amended) The method of claim 4, wherein any content of the other band signals contained in each band signal is attenuated to half of their respective original levels.

34. (Amended) The method of claim 4, wherein any content of the other band signals is completely attenuated from each band signal so as to not be present at all therein.

C11

35. (Amended) The method of claim 5, wherein the audio signal is divided into from 4 to 10 band signals.

36. (Amended) The method of claim 5, wherein the audio signal is divided into from 5 to 8 band signals.

37. (Amended) The method of claim 5, wherein the audio signal is divided into 6 band signals.

C12

39. (Amended) The method of claim 8, wherein the low pass filtering is realized by means of a digital convolution over15 to 25 values.

40. (Amended) The method of claim 8, wherein the low pass filtering is realized by means of a digital convolution over 19 values.

C13

42. (Amended) The method of claim 11, wherein the energy signal is generated by squaring said audio signal or said signal derived therefrom.

43. (Amended) The method of claim 13, wherein the second low pass filtering is effected digitally in the form of a convolution over 40-55 values.

44. (Amended) The method of claim 13, wherein the second low pass filtering is effected digitally in the form of a convolution over approximately 48 values.

C14

48. (Amended) The method of claim 15, wherein the differentiation is performed by computing the difference between two respective values of the energy signal.

Sub E1

C15

60. (New) A method for storing an electric signal representing recorded ambient noise in compressed form, the method comprising:

periodically recording samples of the ambient noise using a sound transducer;

normalizing the amplitude of a signal output of the transducer within a first predetermined range D;

mapping the normalized amplitude values of the sampled ambient noise onto a second predetermined range of values using a non-linear mapping function to obtain an emphasis of selected value ranges within the first and/or second predetermined ranges;

storing the mapped result in an electronic memory in a digital format.

APPENDIX B

VERSION WITH MARKINGS TO SHOW CHANGES MADE
37 C.F.R. § 1.121(b)(iii) AND (c)(ii)

Specification:

Paragraph beginning at page 2, line 24:

[The further claims indicate preferred embodiments, devices for carrying out the method, and applications.]

Claims:

2. (Twice Amended) The method of claim [1] <u>60</u>, wherein said nonlinear function has a slope dW/dD which decreases with increasing values in the range D to obtain an emphasis of the smallest values of [said] <u>the</u> first range of values.

3. (Twice Amended) The method of claim [1] <u>60</u>, wherein [said] <u>the</u> mapped result is represented by binary numbers having a fixed number of binary digits from 3 to 16 bits.

4. (Twice Amended) The method of claim [1] <u>60</u>, further comprising dividing [said] <u>the</u> audio signal into at least two band signals by filtering, with each one of the band signals containing a frequency range of the audio signal, and wherein any content of the other band signals contained in [said] each band signal is present only in an attenuated form.

5. (Twice Amended) The method of claim 4, wherein [said] <u>the</u> audio signal is divided into from 3 to 15 band signals.

6. (Amended) The method of claim 4, wherein [said] <u>the</u> band signals essentially contain frequency ranges of the same width each, and all frequency ranges are comprised in the range of 500 Hz to 10,000 Hz.

8. (Twice Amended) The method of claim 7, wherein [said] <u>the</u> low pass filtering is realized by means of a digital convolution over 10-30 values.

11. (Twice Amended) The method of claim [1] 60, further comprising generating an energy signal which is proportional to an energy content of the ambient noise from [said] the audio signal or from a signal derived from [said] the audio signal.

12. (Amended) The method of claim 11, wherein [said] the energy signal is subjected to a second low pass filtering.

13. (Twice Amended) The method of claim 12, wherein [said] the second low pass filtering is effected digitally in the form of a convolution over 20 to 70 values.

14. (Twice Amended) The method of claim 13, wherein [said] the second low pass filtering is followed by a second data reduction where one energy value among n filtered values is selected, n being at least equal to 2.

16. (Amended Three Times) The method of claim [1] 60, wherein the range of normalized values D is defined by a lower limit D_u, and an upper limit D_o, and wherein the normalization is effected by:

- obtaining the maximum of the absolute value of the audio signal or the derived signal within the duration of normalizing the audio or derived signal, which is shorter than or equal to the duration of a hearing sample,
- multiplying the reciprocal value of said maximum by $(D_0 - D_u + 1)$, and
- multiplying this product by each value of the audio or derived signal within the duration of the normalized signal.

17. (Twice Amended) The method of claim [1] 60, wherein essentially all steps of the method are performed by integer or fixed point arithmetic.

18. (Twice Amended) Device for carrying out the method of claim [1] 60, comprising a hearing sample unit comprising at least one signal processor for performing at least one processing step of the method.

19. (Twice Amended) The device of claim 18, further comprising a non-volatile semiconductor memory connected to [said] the processor for storing the results of the method.

20. (Twice Amended) The device of claim 18, further comprising a timer connected to a power supply of [said] the hearing sample unit for switching off the hearing sample unit when no processing activity is required.

21. (Twice Amended) The device of claim 19, wherein a power supply of said non-volatile memory and/or [said] the memory itself is connected to a timer in such a manner that the memory is essentially capable of being operated only during the storage of the results in order to reduce the energy consumption by the memory.

27. (Twice Amended) A magnetic, optical or magneto-optical data carrier, containing a recorded program which executes the method according to claim [1] 60.

29. (Twice Amended) Device comprising at least one program controlled processor unit and a memory for storing a program controlling [said] the processor unit, wherein [said] the memory contains a program which controls at least one of the operations of the method of claim [1] 60.

30. (Amended) The method of claim [1] 60, wherein the electroacoustic transducer is a microphone.

31. (Amended) The method of claim 3, wherein [said] the mapped result is represented by binary numbers having a fixed number of binary digits from 4 to 8 bits .

32. (Amended) The method of claim 3, wherein [said] <u>the</u> mapped result is represented by binary numbers having 4 bits of binary digits.

33. (Amended) The method of claim 4, wherein any content of the other band signals contained in [said] each band signal is attenuated to half of their respective original levels.

34. (Amended) The method of claim 4, wherein any content of the other band signals is completely attenuated from [said] each band signal so as to not be present at all therein.

35. (Amended) The method of claim 5, wherein [said] <u>the</u> audio signal is divided into from 4 to 10 band signals.

36. (Amended) The method of claim 5, wherein [said] <u>the</u> audio signal is divided into from 5 to 8 band signals.

37. (Amended) The method of claim 5, wherein [said] <u>the</u> audio signal is divided into 6 band signals.

39. (Amended) The method of claim 8, wherein [said] <u>the</u> low pass filtering is realized by means of a digital convolution over15 to 25 values.

40. (Amended) The method of claim 8, wherein [said] <u>the</u> low pass filtering is realized by means of a digital convolution over 19 values.

42. (Amended) The method of claim 11, wherein [said] <u>the</u> energy signal is generated by squaring said audio signal or said signal derived therefrom.

43. (Amended) The method of claim 13, wherein [said] <u>the</u> second low pass filtering is effected digitally in the form of a convolution over 40-55 values.

44. (Amended) The method of claim 13, wherein [said] the second low pass filtering is effected digitally in the form of a convolution over approximately 48 values.

48. (Amended) The method of claim 15, wherein [said] the differentiation is performed by computing the difference between two respective values of the energy signal.

APPENDIX C

COMPLETE SET OF "CLEAN" CLAIMS PURSUANT TO 37 C.F.R. §1.121(C)(3)

60. (New) A method for storing an electric signal representing recorded ambient noise in compressed form, the method comprising:

periodically recording samples of the ambient noise using a sound transducer;

normalizing the amplitude of a signal output of the transducer within a first predetermined range D;

mapping the normalized amplitude values of the sampled ambient noise onto a second predetermined range of values using a non-linear mapping function to obtain an emphasis of selected value ranges within the first and/or second predetermined ranges;

storing the mapped result in an electronic memory in a digital format.

2. (Twice Amended) The method of claim 60, wherein said nonlinear function has a slope dW/dD which decreases with increasing values in the range D to obtain an emphasis of the smallest values of the first range of values.

3. (Twice Amended) The method of claim 60, wherein the mapped result is represented by binary numbers having a fixed number of binary digits from 3 to 16 bits.

4. (Twice Amended) The method of claim 60, further comprising dividing the audio signal into at least two band signals by filtering, with each one of the band signals containing a frequency range of the audio signal, and wherein any content of the other band signals contained in each band signal is present only in an attenuated form.

5. (Twice Amended) The method of claim 4, wherein the audio signal is divided into from 3 to 15 band signals.

6. (Amended) The method of claim 4, wherein the band signals essentially contain frequency ranges of the same width each, and all frequency ranges are comprised in the range of 500 Hz to 10,000 Hz.

7. (Amended) The method of claim 4, wherein the band signals are generated by splitting once or a cascaded multiple of times an input signal which is either the audio signal or an output signal obtained according to the following steps:

- first low pass filtering to generate a first output band signal, and
- subtracting the first output band signal from the input signal to generate a second output band signal.

8. (Twice Amended) The method of claim 7, wherein the low pass filtering is realized by means of a digital convolution over 10-30 values.

9. (Amended) The method of claim 8, wherein for the purpose of the low pass filtering, the convolution is performed using the terms $a_i * x_{t-i}$, wherein the coefficients a_i, $0 < i < 18$, being approximately equal to {0.03, 0.0, -0.05, 0.0, 0.06, 0.0, -0.11, 0.0, 0.32, 0.50, 0.32, 0.0, -0.11, 0.0, 0.06, 0.0, -0.05, 0.0, 0.03}.

10. (Amended) The method of claim 7, wherein the input signal is digitized and only every nth value of each division stage is added to the band signal, n being greater than or equal to 2, in order to compensate for the increased data volume resulting from the splitting into band signals.

11. (Twice Amended) The method of claim 60, further comprising generating an energy signal which is proportional to an energy content of the ambient noise from the audio signal or from a signal derived from the audio signal.

12. (Amended) The method of claim 11, wherein the energy signal is subjected to a second low pass filtering.

13. (Twice Amended) The method of claim 12, wherein the second low pass filtering is effected digitally in the form of a convolution over 20 to 70 values.

14. (Twice Amended) The method of claim 13, wherein the second low pass filtering is followed by a second data reduction where one energy value among n filtered values is selected, n being at least equal to 2.

15. (Amended) The method of claim 11, further comprising performing a subsequent differentiation of the energy signal with respect to time to obtain an energy difference signal.

16. (Amended Three Times) The method of claim 60, wherein the range of normalized values D is defined by a lower limit D_u, and an upper limit D_o, and wherein the normalization is effected by:

- obtaining the maximum of the absolute value of the audio signal or the derived signal within the duration of normalizing the audio or derived signal, which is shorter than or equal to the duration of a hearing sample,
- multiplying the reciprocal value of said maximum by $(D_0 - D_u+1)$, and
- multiplying this product by each value of the audio or derived signal within the duration of the normalized signal.

17. (Twice Amended) The method of claim 60, wherein essentially all steps of the method are performed by integer or fixed point arithmetic.

18. (Twice Amended) Device for carrying out the method of claim 60, comprising a hearing sample unit comprising at least one signal processor for performing at least one processing step of the method.

19. (Twice Amended) The device of claim 18, further comprising a non-volatile semiconductor memory connected to the processor for storing the results of the method.

20. (Twice Amended) The device of claim 18, further comprising a timer connected to a power supply of the hearing sample unit for switching off the hearing sample unit when no processing activity is required.

21. (Twice Amended) The device of claim 19, wherein a power supply of said non-volatile memory and/or the memory itself is connected to a timer in such a manner that the memory is essentially capable of being operated only during the storage of the results in order to reduce the energy consumption by the memory.

22. (Amended) The device of claim 18, wherein the device is an object which is usually carried by persons.

23-26 (Withdrawn from consideration)

27. (Twice Amended) A magnetic, optical or magneto-optical data carrier, containing a recorded program which executes the method according to claim 60.

28. (Amended - Withdrawn from consideration)

29. (Twice Amended) Device comprising at least one program controlled processor unit and a memory for storing a program controlling the processor unit, wherein the memory contains a program which controls at least one of the operations of the method of claim 60.

30. (Amended) The method of claim 60, wherein the electroacoustic transducer is a microphone.

31. (Amended) The method of claim 3, wherein the mapped result is represented by binary numbers having a fixed number of binary digits from 4 to 8 bits .

32. (Amended) The method of claim 3, wherein the mapped result is represented by binary numbers having 4 bits of binary digits.

33. (Amended) The method of claim 4, wherein any content of the other band signals contained in each band signal is attenuated to half of their respective original levels.

34. (Amended) The method of claim 4, wherein any content of the other band signals is completely attenuated from each band signal so as to not be present at all therein.

35. (Amended) The method of claim 5, wherein the audio signal is divided into from 4 to 10 band signals.

36. (Amended) The method of claim 5, wherein the audio signal is divided into from 5 to 8 band signals.

37. (Amended) The method of claim 5, wherein the audio signal is divided into 6 band signals.

38. The method of claim 7, wherein all first low pass filterings have a same Q-factor.

39. (Amended) The method of claim 8, wherein the low pass filtering is realized by means of a digital convolution over15 to 25 values.

40. (Amended) The method of claim 8, wherein the low pass filtering is realized by means of a digital convolution over 19 values.

41. The method of claim 10, wherein n is equal to 2.

42. (Amended) The method of claim 11, wherein the energy signal is generated by squaring said audio signal or said signal derived therefrom.

43. (Amended) The method of claim 13, wherein the second low pass filtering is effected digitally in the form of a convolution over 40-55 values.

55. The method of claim 22, wherein the device is incorporated in the form of a wristwatch.

56-59. (Withdrawn from consideration)

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